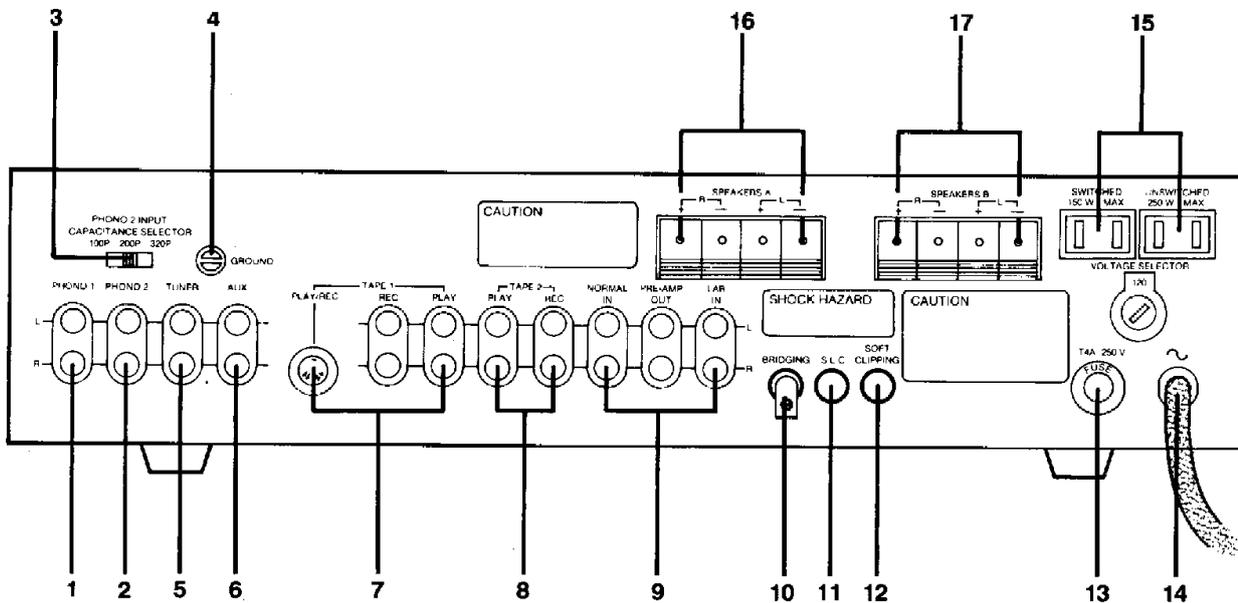
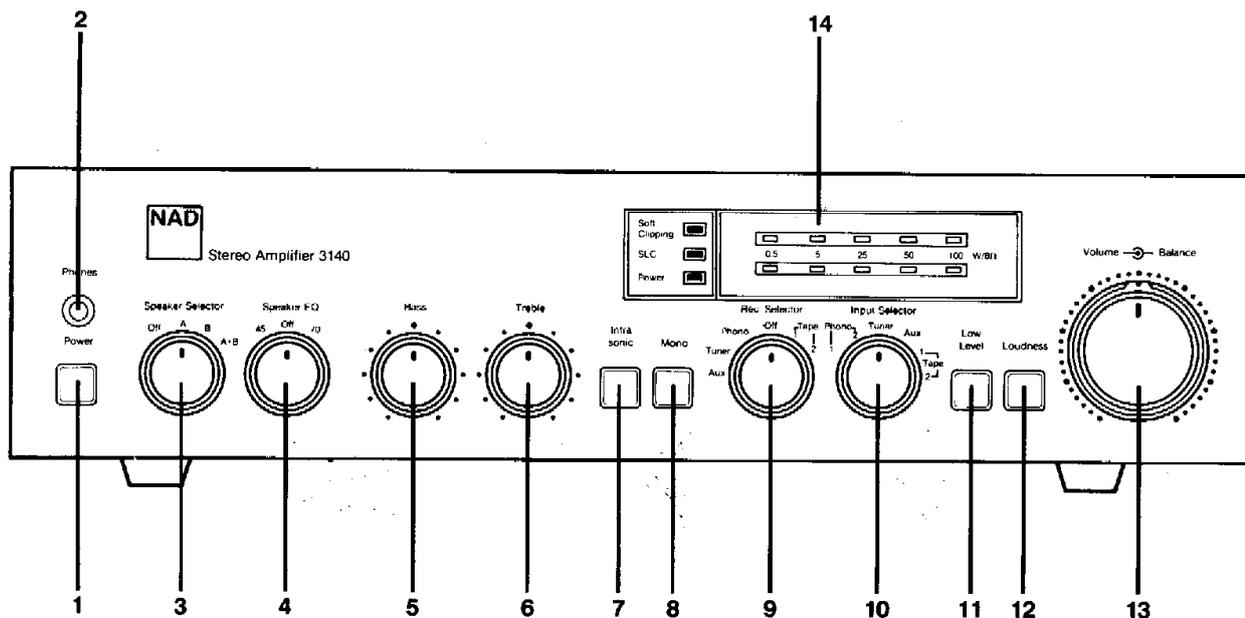


- 1. Phono 1 input
- 2. Phono 2 input
- 3. Capacitance Selector
- 4. Ground terminal
- 5. Tuner input
- 6. Auxiliary input
- 7. Tape 1 input/output
- 8. Tape 2 input/output
- 9. Pre-amp Out, Normal In, Lab In
- 10. Bridging mode switch
- 11. Speaker Lead Compensator
- 12. Soft Clipping switch
- 13. AC Line Fuse
- 14. AC Line Cord
- 15. AC Convenience Outlets
- 16. Speaker Terminals (A)
- 17. Speaker Terminals (B)



- 1. Power
- 2. Phones
- 3. Speaker Selector
- 4. Speaker EQ
- 5. Bass
- 6. Treble
- 7. Infrasonic Filter
- 8. Mono
- 9. Recording Selector
- 10. Input Selector
- 11. Low-Level (audio muting)
- 12. Loudness compensation
- 13. Volume/Balance
- 14. LED Power Meter



REAR PANEL CONNECTIONS

1. PHONO1 INPUT. This input is designed for use with "high-output" moving-coil cartridges such as the NAD 9000, Dynavector (Ultimo) 10A and 10X, Satin M18 and M117, Adcom XC/LT, et al.

This input may also be used with cartridges of other types which have lower-than-normal output voltage, such as the B&O MMC series and Audio-Technica models AT-22 through AT-25. The Phono 1 input has a standard input impedance (47K ohms resistance, 100 pF capacitance) but is 10 dB more sensitive than the Phono 2 input.

Moving-coil pickups with low output voltage should be used with an external transformer or pre-preamp. The output from the step-up device may then be connected to either Phono 1 or Phono 2.

2. CAUTION: Some step-up devices may produce a high enough output level to overload the Phono 1 input, yielding distortion; in case of doubt, use Phono 2. Here's one way to tell: if you find that the amplifier is driven to full power (as indicated by the peak-reading front-panel LED display) with a volume-control setting of 12 o'clock or lower, then you should switch to the less sensitive Phono 2 input. Conversely, if you must turn the volume control up beyond the 3 o'clock position in order to drive the amp to full peak power, then the extra gain of the Phono 1 input will be beneficial.

2. PHONO 2 INPUT. This input is intended for the majority of phono cartridges of the moving magnet, induced magnet, moving flux, and moving iron (variable reluctance) types.

Plug the signal cables from your turntable into these jacks. If the cables or plugs are color-coded, refer to your turntable's instruction manual to learn which cable or plug is for the Left channel and which for the Right. Be careful to insert each plug *fully* into the socket so that the plug's metal skirt fits tightly over the exterior of the socket. If necessary, carefully crimp the plug's metal skirt slightly so as to obtain a tight fit with the socket.

3. CAPACITANCE SELECTOR. This switch selects the input capacitance for PHONO 2 on/y. It enables you to optimize the load capacitance for those cartridges whose frequency response is affected by this parameter.

If you are using a low-inductance pickup (such as a Grado or Micro-Acoustics), or a moving-coil cartridge with a transformer or pre-preamp, then the setting of the Capacitance Selector is unimportant. But with many high-inductance magnetic pickups, the capacitance setting will audibly alter the sound of the pickup.

In order to select the best value of preamp input capacitance you must first determine the *total* capacitance recommended for the cartridge. This will usually be included in the maker's specifications, and may also be mentioned in magazine reviews of the cartridge. Next, subtract the capacitance of your turntable's tone-arm wiring and signal cables. (Check the specifications supplied with the tonearm, or write to the manufacturer of the tonearm, or as a last resort assume a typical value of 150 pF.) After this subtraction, what remains is the desired value of preamp input capacitance. Set the Capacitance Selector to the nearest value. It is not necessary to match the computed value exactly: with most phono cartridges a variation of 50 pF one way or the other will produce only a very slight change in frequency response.

Example: suppose you are using a Stanton 881S pickup cartridge in a Pioneer turntable. Stanton specifies a recommended load capacitance of 275 pF for the cartridge, and the Pioneer turntable has a cable capacitance of about 100 pF. 275 minus 100 equals 175 pF, so you should set the Capacitance Selector to the nearest value, 200 pF.

If you prefer, you may simply set the Capacitance Selector by ear while listening to recordings which are strong in high-frequency overtones. Typically, when the capacitance is too low the upper-midrange (the soprano voice range) will be softened and the response at the highest frequencies will be peaky, leading to edgy

violin tone and increased surface noise. Too high a value of capacitance will bring the upper-midrange forward while rolling off the extreme highs.

4. GROUND. If your turntable is equipped with a grounding wire (usually a green wire terminating in a U-shaped spade lug), connect it to this terminal. Turn the thumb-nut counter-clockwise, place the spade lug under the nut, and tighten the thumb-nut clockwise to secure the lug. If the grounding wire has no terminal lug, strip off a half-inch (1 to 2 cm) of insulation to expose the bare wire, twist the wire strands tightly together, insert the wire through the small hole in the shaft of the Ground terminal, and tighten the thumb-nut to fasten the wire in place.

5. TUNER. Connect the signal cable from a radio tuner to these jacks. As with all of the other input/output jacks, the upper one in each pair is for the left channel and the lower jack is for the right channel.

6. AUX. These auxiliary input jacks are for any "line level" signal source—such as a spare tape player, a television sound tuner, the audio line output from a videocassette or videodisc player, or a child's record player containing a high-output ceramic pickup cartridge.

7. TAPE 1 REC/PLAY. Two types of connectors are provided for use with a stereo tape recorder: separate pairs of RECORD and PLAY phono jacks, and a five-pin DIN socket. If your recorder has only DIN-type plugs, use the DIN connector. If your recorder has both a DIN plug and pairs of phono jacks, it is preferable to use the phono plug connections. (Do not use both the phono plugs and the DIN plug simultaneously.)

The TAPE 1 connections may be used with tape recorders of all types: cassette, micro-cassette, open-reel, eight-track, digital, etc. To make recordings, connect a stereo patch cord from the RECORD jacks to the LINE or RADIO input jacks on the recorder (*not* to its microphone inputs). To play back tapes, connect a stereo patch cord from the recorder's LINE output jacks to the amplifier's PLAY input jacks.

8. TAPE 2 REC/PLAY. These jacks enable connection of a second tape recorder of any type, and the amplifier is wired to permit copying tapes from one recorder to the other. Connect a cable from the RECORD jacks to the tape deck's LINE or RADIO input jacks, and another cable from the amplifier's PLAY jacks to the recorder's LINE outputs. The upper jack is for the left channel and the lower jack for the right channel.

The TAPE 2 jacks may be used for a signal-processing accessory instead of a second tape machine. Examples of such accessories include a dynamic range processor, a dynamic noise filter, an impulse noise ("tick and pop") suppressor, and any other processor whose operation depends on the setting of a signal threshold. Connect a patch cord from the REC jacks to the processor's inputs, and another patch cord from the PLAY jacks to the processor's outputs.

Other signal processors, such as a graphic equalizer or the special equalizer supplied for use with some loudspeakers (e.g. Bose, Electro-Voice, KLM), may be connected either to the tape jacks or at the output of the preamp. The choice is a matter of convenience.

9. PRE-AMP OUT, NORMAL IN, LAB IN. Each channel of the amplifier is comprised of two independent sections or stages: the control preamplifier (including the phono preamp and most front-panel controls), and the power amplifier (which provides the power to drive loudspeakers). In normal operation the preamp and power amp are connected together via U-shaped metal jumpers; check to be sure that they are fully inserted into the jacks and that nothing is touching them.

Two sets of power amp inputs are provided. The LAB inputs have wideband frequency response extending uniformly from low infrasonic to high ultrasonic frequencies, and may be used for laboratory tests and special applications. The NORMAL inputs are equipped with infrasonic and ultrasonic filters; these reject any

interference which occurs outside of the audible frequency range, in order to prevent intermodulation distortion and preserve the amplifier's power for music.

For conventional operation the PRE-AMP OUTjacks are connected to the NORMAL IN jacks by means of the metal jumpers. Removal of the jumpers (with the POWER switched OFF) enables various signal-processing accessories to be connected in the signal path between preamp and power amp: an equalizer, a time-delay ambience reproducer, a stereo image enhancer, etc. To use a signal processor, connect a stereo patch cord from the PREAMP OUT jacks to the processor's line-level input jacks, and a second patch cord from the processor's output jacks to the amplifier's NORMAL IN jacks. (Note: any signal processor whose operation depends on the setting of a threshold, such as a dynamic noise filter or a DBX decoder, should be connected to one of the sets of TAPE REC/PLAY jacks—where the signal levels are unaffected by volume and tone controls—rather than to the PRE-AMP OUTjacks.)

If you remove the metal jumpers, save them in case you may want to disconnect the signal processor and return to normal operation at a later time. If the jumpers should be lost, a conventional stereo patch cord can be used to connect PRE-AMP OUT to NORMAL IN or LAB IN.

The NAD 3140 can be used as the heart of an elaborate audiophile sound System. For example the PRE-AMP OUT jacks may be connected via a stereo patch cord to any high-quality separate power amplifier. To use a separate high-power amplifier for one set of loudspeakers while continuing to use the NAD 3140's built-in power amp for headphones or another set of loudspeakers, simply install Y-connector adapters to split the signal from the PRE-AMP OUTjacks. The preamp stage is capable of driving several power amplifiers in parallel, or driving the long signal cables required to connect to power amps which are located near the speakers, or to "powered" loudspeakers with built-in power amps. The preamp output can also be fed to a time-delay ambience system, with the 3140's built-in power amp used to drive either the main stereo speakers or the time-delayed ambience speakers. In a bi-amplified System the preamp output is fed to the input of an electronic crossover; the low-frequency output of the crossover unit is fed to the amplifier which drives the woofers, while the high-frequency output of the crossover unit is fed to the 3140's NORMAL IN or LAB IN jacks, and the speakers' tweeters are connected directly to the 3140's SPEAKER terminals.

10. BRIDGING MODE SWITCH. This switch "bridges" the NAD 3140's two power amplifier channels to form a monophonic amplifier with more than double the output power. To convert to bridged operation, the following procedure should be followed.

(1) Disconnect the metal jumper or signal cable from the left-channel input to the power amplifier section of the 3140. In the bridged (monophonic) mode the amplifier is driven through its right-channel input only (either the NORMAL or LAB input socket). If you need stereophony reproduction, the left-channel PRE-AMP OUT signal can be used to drive a second, separate power amplifier and loudspeaker, (If another 3140 in bridged mode is used for the second channel, it too will be driven through its right channel power amp input regardless of whether it is used for the right or left speaker.)

(2) Disconnect any speaker wires from both the SPEAKERS A and SPEAKERS B terminals. From the speaker which is to be driven by the 3140, connect its positive lead to the red R+ terminal, and its negative lead to the red L+ terminal, in the SPEAKERS A group of terminals. NOTE; do *not* connect any wires to the black (R- and L-) terminals.

If you want to drive two speakers in parallel, connect the second speaker's leads to the red (R+ and L+) terminals in the SPEAKERS B group; make no connection to the black (R- and L-) terminals. NOTE: in the bridged monophonic mode, the loudspeaker's impedance appears to be halved as "seen" by the amplifier. An 8-ohm load looks like 4 ohms, a 4-ohm load looks like

2 ohms, and a pair of 4-ohm speakers operated in parallel will look like a 1-ohm load. Driving a pair of such speakers to high levels may cause the amplifier to overheat, activating its protection circuits.

CAUTION: In the bridged mode the speaker wires must be "floating" with respect to ground. Do NOT connect the speaker wires to anything which shares a common ground between stereo channels (such as a headphone adapter) nor a common ground with the 3140's inputs (such as a switching comparator or a distortion analyzer).

(3) After the preceding conditions have been satisfied, use the bridging switch. It is normally covered by a metal bracket to prevent its accidental use. Loosen the screw which holds the bracket in place and swing it out of the way or remove it altogether. Press and release the BRIDGING button. The button has two settings:

IN—normal stéréo mode

OUT—bridged monophonic mode

To return the 3140 to normal stereo operation at a later date, simply depress the BRIDGING button to engage the normal stereo mode, fasten the metal bracket in place over the button to prevent accidental re-setting of the switch, feed stereo signals into the left and right channel power amp inputs, and connect loudspeakers to the appropriate terminals as described below under SPEAKERS A and SPEAKERS B.

11. SPEAKER LEAD COMPENSATOR (SLC™). When a speaker is connected to an amplifier through a length of wire, the resistance of the wire will affect the signal reaching the speaker terminals. If the wire is sufficiently short or sufficiently large in diameter, its resistance will be negligible. As a general rule, 18-gauge or heavier wire should be used for lengths of up to 20 feet (6 meters) and 16-gauge or heavier wire for lengths of up to 30 feet (10 meters). But if you use thinner wire, or greater lengths, the wire resistance may have a directly audible effect on the speaker's sound—especially with low-impedance speakers, or pairs of speakers wired in parallel. This effect is of four kinds:

(1) Some power is dissipated in the wires, and the signal delivered to the speaker is slightly reduced in level.

(2) Since the speaker's impedance varies with frequency, the reduction in signal level varies in proportion; i.e. the tonal balance of the signal is altered.

(3) Typical speaker impedances are complex, varying with signal level (for example, the voice-coil inductance varies as the coil moves in and out of the magnet gap), and may become non-linear at high volume levels. The resulting non-linear current flow produces a non-linear (i.e. distorted) voltage across the resistance of the speaker leads. Thus the audio signal may be completely distortion-free at the amplifier's output terminals, yet exhibit several percent of distortion at the far end of the leads where they connect to the speaker terminals.

(4) Finally, the wire resistance reduces the amplifier's damping factor.

The Speaker Lead Compensator (SLC) cancels the effects of the wire resistance, eliminating the distortion and restoring the performance which would be obtained if the wire had no resistance. The SLC is calibrated for a specific amount of wire resistance, corresponding to the following lengths of standard wire sizes:

GAUGE	FEET	METERS
1	97	30
4	61	19
16	38	12
18	24	7
20	15	5
22	1	4
24	2	3
26	10	2

Compare your speaker wire size and length to these figures. (If your speaker leads are of differing lengths in the two channels, use the average.) If your speaker leads are less than half of the length specified above, leave the SIC switched off. If your speaker leads are more than half of the length in the table, depress the SLC button to engage the Speaker Lead Compensator. The more nearly your speaker leads match the length in the table, the more precise will be the SLC's cancellation of the effects of the wire resistance.

Example: if your speaker connecting leads are 18-gauge wires and are less than 19 feet in length from amp to speaker, leave the SLC switched off (button DUT). If the wires are longer than 19 feet, switch the SLC on (button IN).

12. SOFT CLIPPING™ When an amplifier is overdriven beyond its specified power output it normally produces "hard clip-ping" of the signal with harsh distortion and power-supply buzz as the output transistors saturate. The NAD Soft Clipping circuit gently limits the output waveform and minimizes audible distortion when the amplifier is overdriven. If your listening involves only relatively low peak power levels (as shown by the front panel LED display), the Soft Clipping circuit may be left off (button DUT). But in general we recommend that it be switched on (button depressed), especially when playing music containing high peak levels.

13. FUSE. The fuse protects the amplifier from damage in case of internal parts failure. If the fuse blows, the amplifier will not operate and the front-panel LED display will not illuminate.

If the fuse blows, unscrew the fuse holder and install a replacement fuse of the same size and type (a 4-ampere slow-blow fuse in areas where the AC line voltage is 110 or 120 volts, a 2-amp fuse where the power line is 220 or 240 volts). CAUTION: unplug the AC line cord before changing the fuse.

14. AC LINE CORD and VOLTAGE SELECTOR. The AC line cord should be plugged into a "live" wall socket. The NAD 3140 will operate with AC line voltages from 100 to 240 volts, and a line frequency of either 50 or 60 Hz. Use a small coin or screwdriver blade to adjust the Voltage Selector to match the AC powerline voltage in your area. (Normally this will be done by the NAD distributor before the amplifier is delivered to your retailer.) If you travel overseas you can reset the Voltage Selector for each country.

15. AC CONVENIENCE OUTLETS. (Not in U.K. model.) The AC line cords of other stereo components may be plugged into these accessory outlets. The SWITCHED outlet is intended for all-electronic products (e.g. a radio tuner, equalizer, or other signal processor), and it will be switched on and off by the main POWER button on the front panel. The UNSWITCHED outlet is intended to power any product involving mechanical operations (e.g. a turn-table or tape deck); such products should be turned on and off with their own power switches.

16. SPEAKERS A. If the wiring to each speaker will be no longer than about 20 feet (6 meters), then connections should be made using 18-gauge wire such as common lamp cord ("Zip" cord), available from hardware and electrical-supply stores in either white, brown, or black insulation. If the wiring to the speakers will be longer than about 20 feet, heavier 16-gauge Zip cord is pre-furred. The use of heavy-duty wiring is especially desirable if you are using speakers of low impedance or two pairs of speakers wired in parallel. (If you will be using a thinner wire size or un-usually long wires to the speakers, then you should use the Speaker Lead Compensator as described in section 11.)

To make connections, separate the two conductors of the cord, strip off about a half-inch (1 cm) of insulation from each, and in each conductor twist together the exposed wire strands. Fully depress the colored tab below each terminal in order to open up the small hole in the terminal, insert the bared wire into the hole, and release the tab; the terminal will grasp the wire and hold it in

place. Repeat for each conductor, connecting the wires from the left-channel speaker to the (L+) and (L-) terminals and the wires from the right-channel speaker to the (R+) and (R-) terminals in the SPEAKERS A group. Check to be sure that no loose strand of wire is touching any adjacent terminal.

PHASING. Stereo speakers should operate in phase with each other in order to yield a good stereo image and to reinforce rather than cancel each other's output at low frequencies. If your speakers are easily moved, phasing can easily be checked. Make the connections to the speakers, place the speakers face-to-face only a few inches apart, play some music, and listen. Then swap the connection of the two wires at the back of one of the speakers, and listen again. The connection which produces the fullest, boomiest bass output is the correct one. Connect the wires securely to the speaker terminals, being careful to avoid leaving loose strands of wire which might touch the wrong terminal and create a partial short-circuit, and then move the speakers to their intended locations.

If the speakers cannot easily be set face-to-face, then phasing must rely on the "polarity" of the connecting wires. Note that the SPEAKERS terminals on the amplifier are color-coded: in each channel the terminal with the red tab has positive V polarity and the black terminal is negative "-". The terminals at the rear of the speakers are also marked for polarity, either via red and black connectors or by labels: "+", 8n, or 1 for positive, "-", G, or O for negative. As a general rule the positive (red) terminal on the amplifier is to be connected to the positive terminal of the speaker, in each channel. To facilitate this, the two conductors comprising the speaker wire in each channel are different, either in the color of the wire itself (copper vs. silver) or in the presence of a small ridge or rib pattern on the insulation of one conductor. Use this pattern to establish *consistent* wiring to both speakers of a stereo pair. Thus if you connect the copper-colored wire (or ribbed insulation) to the red amplifier terminal in the left channel, do the same in the right channel. And at the other end of the wire, if you connect the copper-colored wire (or the ribbed insulation) to the red or positive terminal on the left-channel speaker, do the same at the right-channel speaker.

17. SPEAKERS B. A second pair of loudspeakers may be connected to the 3140, using these terminals, in the same manner as the speakers connected to the SPEAKERS A terminals.

If the second pair of speakers is located near the first pair in the same room, then they must be correctly phased with respect to the first pair as well as with each other. But if the second pair of speakers is located away from the first pair—in another room for example—then their phasing need not be consistent with that of the first pair. (Of course, as with any stereo pair of speakers, the extension speakers still must be in phase with each other.)

The SPEAKERS B terminals may also be used to connect an adapter unit for electrostatic headphones.

Another useful option for the SPEAKERS B terminals is to connect a second pair of speakers wired for "ambience recovery," enhancing the apparent spaciousness of stereo recordings. Locate a pair of small loudspeakers in the rear corners of the room, slightly behind the main listening area and as far as possible to the left and right. Connect a wire from the (L+) terminal to the "positive" or 8-ohm input terminal of the left-rear speaker, and a wire from the (R+) terminal to the "positive" terminal of the right-rear speaker. Make no connection to the (L-) and (R-) terminals; instead, connect a wire from the "negative" or "ground" terminal of the left-rear speaker to the same terminal of the right-rear speaker. Thus wired, these rear speakers receive the left-minus-right "difference" portion of the stereo signal.

*SLC and Soft Clipping are trademarks of NAD (USA), Inc. Registration pending.

FRONT PANEL CONTROLS

1. POWER. Depress to switch on the amplifier and any other equipment plugged into the SWITCHED convenience outlet on the rear panel. To switch off the power, depress the button again and release it.

If you prefer, you may leave the 3140's POWER switch permanently engaged and use an external switch (such as a clock timer) to turn the power on and off.

2. PHONES. Plug stereo headphones in here. The circuit will provide proper drive signals for all conventional stereo headphones regardless of their impedance, with just one exception: electrostatic headphones usually are supplied with an adapter unit which must be connected directly to the speaker terminals on the rear panel.

Before plugging conventional headphones into the PHONES Jack, turn down the VOLUME control for safety. And when you are not listening to the headphones it is wise to unplug them from the PHONES jack. Otherwise, when listening to loudspeakers you might turn up the volume to a level which would feed excessively strong signals to the headphones and damage them.

You may freely use headphone extension cables. If you want to use a headphone Y-connector to drive two headsets simultaneously, they should be identical models. Connecting together two headphones which differ widely in impedance usually will produce a substantial loss of volume in the headset having the higher impedance (or in both).

3. SPEAKER SELECTOR. When this switch is set to "A", sound is heard only from the speakers connected to the SPEAKERS A terminals on the rear panel of the 3140. When the switch is set to "B" the SPEAKERS A terminals are shut off and sound is heard only from the speakers connected to the SPEAKERS B terminals. At the "A+B" setting the amplifier's output power is fed to both sets of speakers; they are wired in parallel by the switch. At the "OFF" setting, both sets of speakers are shut off.

Thus if you have your main stereo speakers wired to the "A" terminals and a set of extension speakers wired to the "B" terminals, you can choose to hear only the main speakers (A), only the extension speakers (B), or you can activate both (A+B).

The amplifier's output signal is present at the PHONES jack at all settings of the SPEAKER SELECTOR switch. When using headphones it normally is advisable to switch OFF the loudspeakers; then the VOLUME control may freely be used to adjust the loudness level in the headphones with no fear of overdriving the speakers or disturbing neighbors.

If you have connected the adapter unit for electrostatic headphones to the SPEAKERS B terminals, you can use the SPEAKER SELECTOR to switch between your main stereo speakers (A) and the headphones (B).

If you have connected speakers wired for "ambience recovery" to the SPEAKERS B terminals, you can use the SPEAKER SELECTOR to listen to conventional stereo (A), to switch off the main speakers and listen to only the stereo L-minus-R "difference" signal in the rear speakers (B), or to listen to spatially-enhanced stereo (A+B). You will find that the stereo difference signal is usually lacking in bass. If the difference signal is very weak, the recording lacks stereo separation.

4. SPEAKER EQ. The lowest octaves of deep bass sound are seldom experienced at their full level in stereo playback, for three reasons:

(1) Most loudspeakers are designed for uniform response down to a planned System resonance frequency (usually in the 40-70 Hz range) and roll off rapidly in response below that frequency.

(2) The low bass is often rolled off by filters when records are made in order to limit groove-modulation levels and increase playing times.

(3) Standing waves bias the distribution of low-frequency

energy in listening rooms, weakening the low bass and reinforcing the mid-bass at typical listening positions.

The NAD SPEAKER EQUALIZER compensates for these losses by providing a 12 dB/octave boost below either 45 Hz or 70 Hz. Since all closed-box (acoustic-suspension and infinite-baffle) loudspeaker Systems roll off in response at 12 dB/octave below the woofer/cabinet resonance, the NAD SPEAKER EQUALIZER precisely compensates this rolloff and extends the useful response of the speaker a full octave lower in frequency. Note: while the SPEAKER EQ can be used with "vented" speakers (bass-reflex, tuned-port, auxiliary bass radiator, et al.), these designs usually exhibit a much more rapid rolloff below the system's planned cutoff (typically either 18 or 24 dB/octave); consequently the SPEAKER EQ will not produce the same dramatic benefit with these designs as it does with acoustic-suspension Systems.

If you have full-size speakers with strong response down to the 40-50 Hz range, the 45 Hz setting of the SPEAKER EQ is theoretically "correct." Nevertheless you should feel free to use the 70 Hz setting (instead of the BASS control) to bring up the rolled-off bass in recordings to a satisfying level; let your ears be your guide.

Three CAUTIONS should be observed in using the SPEAKER EQ:

(1) This circuit is intended for use with loudspeakers having woofers eight inches (20 cm) or larger in diameter, preferably those with "long-throw" voice-coils and acoustic-suspension enclosures. It is not recommended for use with small "mini" speakers having woofers smaller than six inches; in most cases they are not designed to accept high power input at low frequencies and will only distort or suffer damage as a result.

(2) Be prepared to reduce or switch off the equalization when playing recordings (especially digitally-mastered ones) which contain unusually potent recorded bass. The SPEAKER EQ boosts deep bass levels by 12 dB (i.e. by a factor of 16 in power). A bass-heavy input signal may Overdrive the amplifier into clipping and Overdrive your woofers beyond their safe excursion limits causing the voice-coils to clatter against their magnet back-plates. As long as it sounds good it probably is OK; but distorted or unmusical sounds are a sign of distress in a woofer.

(3) We recommend that you use the Infrasonic Filter, in order to avoid amplifying inaudible frequencies below 20 Hz, and to preserve the amplifier power and available woofer excursion capacity for the genuine bass energy in the music.

5. BASS. The Bass control adjusts the relative level of the low frequencies in the sound. The electrical response of the amplifier is flattest when the control is set in the detent at the 12 o'clock position. Rotation of the knob to the right (clockwise) increases the level of low-frequency sounds, and rotation counter-clockwise decreases their level. Adjust it to achieve the tonal balance which sounds most natural to you. You will note that at moderate rotations the effect of the Bass control usually is subtle because its action is confined to the lowest audible frequencies. Only at large rotations away from center is there a substantial boost or cut at the mid-bass frequencies which are prevalent in music.

6. TREBLE. The Treble control adjusts the relative level of the high frequencies in the sound. The electrical response of the amplifier is flattest when the control is set in the detent at the 12 o'clock position. Rotation of the knob to the right (clockwise) increases the level of high-frequency sounds, and rotation counter-clockwise decreases their level. Adjust it to achieve the tonal balance which sounds most natural to you. You will note that boosting the Treble increases the brilliance and clarity of details in the sound, but also makes any noise more prominent. Cutting the treble makes the sound mellower and suppresses hiss and record surface noise, but too much Treble cut will make the sound dull.

7. INFRASONIC FILTER. The signal from a record player usually contains strong infrasonic energy (due to disc warps, tone arm resonance, and vibrations reaching the turntable) which, if amplified at full strength, will waste amplifier power and produce excessive woofer cone excursions, muddying the sound. This

button engages an infrasonic filter to attenuate these unwanted signals below 20 Hz. (Another infrasonic filter, unaffected by this switch, is included at the NOF.MAL input to the power amplifier.)

8. MONO. Depressing this button combines the two stereo channels together to produce monophonic sound, minimizing vertical rumble and surface noise when listening to old monophonic records. The button must be OUI for normal stereo listening.

9. RECORDING SELECTOR. This rotary switch selects which input signal will be fed out to the rear-panel TAPE jacks for recording. The selected signal is fed to both TAPE 1 and TAPE 2 and may be recorded simultaneously on two tape machines. The RECORDING SELECTOR operates independently of the INPUT SELECTOR; thus you can record the signal from one program source while listening to a completely separate signal source—i.e. record from the TUNER input while listening to records via either PHONO input, or copy recordings from TAPE 1 onto TAPE 2 while listening to PHONO, TUNER, or an AUX input signal. There is just one limitation to this freedom: when the RECORDING SELECTOR is set to PHONO to tape the signal from a record player, it is the PHONO 2 input signal which is fed to the Tape jacks. If you want to record from the PHONO 1 input, you must set the INPUT SELECTOR to PHONO 1 and cannot simultaneously listen to a different signal source.

When the RECORDING SELECTOR is set to OFF, the signal chosen by the INPUT SELECTOR is fed to the tape jacks for recording. There is an advantage to this: before going to the TAPE jacks the input signal passes through the INFRASONIC FILTER (if it is engaged) and a buffer stage with a low-impedance output which can drive long connecting cables without difficulty.

Signals chosen by the RECORDING SELECTOR pass directly to the tape jacks without going through any intervening electronics. We suggest that when you are not making a recording, set the RECORDING SELECTOR to OFF, thus isolating the tape outputs via a buffer from the main signal path through the amplifier. This ensures that the non-linear input impedance, which some tape recorders exhibit when switched off, will not affect the signals passing through the 3140.

To dub (copy) tapes from TAPE 1 onto TAPE 2, simply set the REC. SELECTOR to TAPE 1. The TAPE 1 playback signal will be fed to the TAPE 2 REC jacks for recording. Then you can set the INPUT SELECTOR to TAPE 1 (to hear the source tape), to TAPE 2 (to monitor the output of the copying recorder) or to PHONO or TUNER if you want to listen to something else while the copying proceeds. Changing the setting of the INPUT SELECTOR will have no effect on the signal fed to the tape recorder by the REC. SELECTOR switch.

Similarly, tapes can be copied from TAPE 2 back to TAPE 1 simply by setting the REC. SELECTOR to TAPE 2.

If you have a signal processor such as an equalizer or a DBX processor connected to the TAPE 2 jacks, you can use it to process the playback signal from the TAPE 1 recorder by setting the REC. SELECTOR to TAPE 1. Then set the INPUT SELECTOR to TAPE 1 to hear the unprocessed signal, or to TAPE 2 to hear the processed signal.

If your turntable is connected to the PHONO 2 inputs, you can use a DBX processor (connected to the TAPE 2 jacks) to decode DBX encoded records. Set the REC. SELECTOR to PHONO; the preamplified PHONO 2 signal will be fed to the DBX processor via the TAPE REC jacks. Then set the INPUT SELECTOR to TAPE 2 to hear the decoded signal.

If your turntable is connected to PHONO 1 and you want to use a DBX processor to play encoded discs, you must either reconnect the turntable to PHONO 2 or connect the DBX decoder to the PRE-AMP OUT and power amp input jacks.

If you want to use an equalizer, DBX processor, or other device to alter the signal being fed to a tape deck for recording, disconnect the recorder from the TAPE 1 jacks and connect it to the processor's own taping jacks.

10. INPUT SELECTOR. This rotary switch selects the signal which you will hear, regardless of the setting of the RECORDING SELECTOR.

If the REC. SELECTOR is set to OFF, the signal chosen by the INPUT SELECTOR will also be fed to the TAPE jacks for recording or other processing.

If you have a three-head tape recorder and wish to monitor its playback output while a recording is being made, use the REC. SELECTOR to feed the desired input signal to the recorder. Then set the INPUT SELECTOR to TAPE 1 or 2 (as appropriate) to hear the monitor output of the recorder.

Similarly, if you have a dynamic processor (such as a DBX decoder) or other signal processor connected to the TAPE 2 jacks and want to hear the processed signal, use the REC. SELECTOR to feed the desired input signal to the processor, and set the INPUT SELECTOR to TAPE 2 to hear the processed signal. Alternatively, if the processor's operation does not involve the setting of a signal threshold, a simpler approach is to connect the processor between the preamp and power amplifier sections of the 3140 (using the PRE-AMP OUT and either the NORMAL IN or LAB IN jacks), leaving the TAPE jacks free for use only with recorders.

11. LOWLEVEL. Pressing this button reduces the volume of the amplified sound by approximately 20 decibels (but has no effect on the signal fed to the REC jacks for taping). It has several practical uses:

- It extends the useful range of the Volume control. With high-output signal sources, with some sensitive medium impedance headphones, or with efficient loudspeakers, you may find that the sound is too loud over most of the range of the Volume control. I.e., you are restricted to using only settings near the lower end of the control range. The use of the LOW LEVEL button makes the full range of the Volume control available to you for normal listening.

- It provides optimum signal/noise ratio for low-level listening in quiet environments. For example, if you are listening to soft music late at night when the surroundings are quiet, the LOW LEVEL button minimizes the already-low residual noise of the preamp and tone control circuits, ensuring that it will never be heard.

- It provides a temporary cut in volume, to be used while answering the telephone for instance. When it is pressed again and released, it restores the volume precisely to the pre-set level.

12. LOUDNESS Pressing this button engages a "loudness compensation" circuit which, at low-to-medium settings of the Volume control, boosts the bass and treble response of the amplifier. This is to compensate for the human ear's reduced sensitivity to low-frequency sounds at low loudness levels, and for the "masking" of high-frequency details by environmental noise. The LOUDNESS function should be disengaged when you are listening to music at life-like volume levels. And at low levels a more accurate, if less convenient, loudness compensation may be obtained by boosting the Bass control.

13. VOLUME/BALANCE. These controls are concentric. The protruding knob is the VOLUME control. The outer ring is the BALANCE control.

The VOLUME control adjusts the overall loudness level of the sound, together with the LOW LEVEL button. It has no effect on the level of the signals fed to the taping jacks. The control is designed for accurate tracking of its two channels, so that the stereo balance will not shift noticeably as the VOLUME control setting is varied,

The BALANCE control adjusts the *relative* levels of the left and right channels. A detent at the 12 o'clock position marks the point of equal balance. Rotation of the ring to the right (clockwise) de-creases the level of the left channel so that only the right channel is heard, thus shifting the sonic image to the right. Rotation of the ring to the left shifts the sonic image toward the left speaker.

Ideally the detented center position of the BALANCE control will be the normal setting. But several common circumstances may cause unequal balance, requiring a compensatory off-center BAL-

ANCE setting to restore the most uniform spread of stereo sound between the speakers. These include unequal output from the two channels of the phono cartridge, differing acoustical environments around the two speakers, or simply a listening position which is closer to one speaker than to the other. Adjust the BALANCE control to produce a natural spread of sound across the space between the speakers, with any monophonic sound (such as a radio announcer's voice) appearing as a phantom image centered midway between them.

14. LED POWER METER. This double row of five LEDs in each channel continually indicates the peak power level which the amplifier is delivering to the loudspeakers. The calibrated level varies from 0.5 watt to 100 watts into an 8-ohm impedance. With a 4-ohm impedance the nominal power is double the indicated value, so the LEDs range from 1 to 200 watts in each channel.

In addition to providing general information on power levels, the LED Power meter also tells you when you should be using the Soft Clipping circuit. If you find that only the first two or three LEDs ever illuminate when you are playing music, then you may leave the Soft Clipping switched OFF. But whenever you find that you are causing all five LEDs to illuminate, even if only momentarily during the highest musical peaks, then you should switch ON the Soft Clipping on the rear panel in order to minimize any harshness or distortion which would occur when the amplifier is overdriven beyond its rated power.

Note that, at the left edge of the Power Meter, there are three "status" LEDs. The lowest of the three is illuminated as a pilot light when the 3140 is on. The middle LED illuminates when the Speaker Lead Compensator (SLC) is engaged. The top LED illuminates to indicate that the Soft Clipping circuit is engaged.

A note on protection. Because the 3140 sounds so clean and musical when driven beyond its nominal power rating and when used to drive low-impedance loudspeakers, you may be tempted to stress it beyond its design capacity. For example it can safely and cleanly drive a 2-ohm impedance with wide-range musical signals whose peak level is several tens of watts and whose average level is much lower; but it will overheat if called upon to deliver high power *continuously* into a low impedance. There are thermostatic circuit breakers in the output stage, which are activated if the output transistors become dangerously hot. When this occurs in either channel the output stage automatically shuts down to protect itself.

Thus if one or both channels of sound go silent while the front-panel LEDs remain illuminated (indicating that the main power-supply fuses and operating voltages are still normal), the thermostatic circuit breakers may have been activated. To resume operation simply turn down the volume and wait a minute or so for the output stage to cool and the circuit breakers to automatically re-set. If the protective circuit breakers interrupt the sound repeatedly, examine the speaker wiring for a possible loose strand of wire causing a partial short-circuit, or reduce the volume level slightly.